Trunks

User Guide

Schmooze Com Inc.

Chapters

Overview

The Trunks module is where you control connectivity to the PSTN and your VoIP provider(s). This is where you also control to interconnect other PBX's for multi-site applications. The most common trunks are SIP and DAHDi (or Zap). Other than the Extensions module, the Trunks module is one of the most critical modules on the system and allows for a great deal of flexibility.

Logging In

- Log into the Trunks module and you should see a screen like this.

Add a Trunk

- Add SIP Trunk
- Add DAHDi Trunk
- Add Zap Trunk (DAHDi compatibility mode)
- Add IAX2 Trunk
- Add ENUM Trunk
- Add DUNDi Trunk
- Add Custom Trunk

Add Trunk
FreePBXTrunk1 (sip)
FreePBXTrunk2 (sip)
Skype (sip)
teliax (sip)
ToRemote (sip)
vitel-inbound (sip)
vitel-outbound (sip)
Channel g0 (zap)

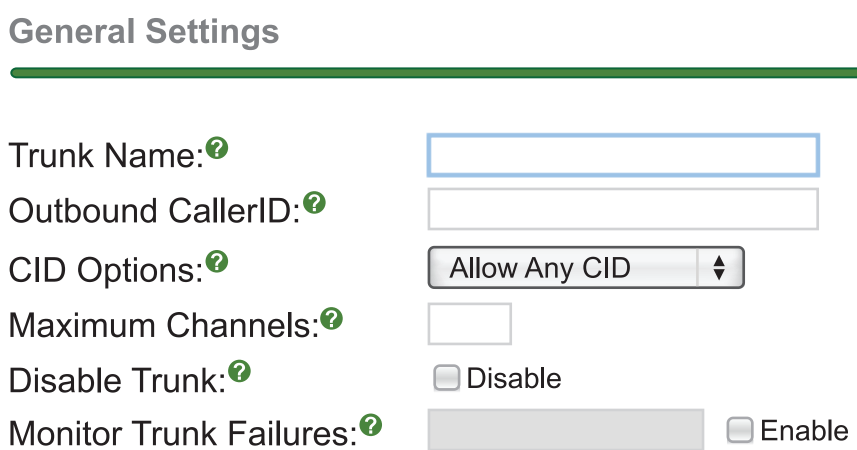
Adding a SIP Trunk

SIP (Session Initiation Protocol) trunks are the most commonly used method of bringing outside dial tone to your system and are relatively easy to set up. However, it should be noted that misconfiguration can lead to toll fraud and care must be taken, especially with regards to "context." There are five core areas displayed for a new SIP trunk: **General Settings**, **Dialed Number Manipulation Rules**, **Outgoing Settings**, **Incoming Settings** and **Registration**. There are examples for a few providers at the end of the guide and see **https://wiki.asterisk.org/wiki/display/AST/Home** for more info.

- **General Settings**:
  - **Trunk Name**- Set a descriptive name for the trunk.
  - **Outbound CallerID**- Use this field to specify caller ID for calls placed out of this trunk with the <NXXNXXXXXX> format. You can also use the format: "hidden" <NXXNXXXXXX> to hide the caller ID sent out over digital lines, if supported (E1/T1/J1/BRI/SIP/IAX2).
  - **CID Options**- This setting determines what CIDs will be allowed out of this trunk. Please **NOTE** that Emergency CIDs defined on an extension or device will **ALWAYS** be used if this trunk is part of an emergency route regardless of these settings.

- **Allow Any CID**: All CIDs, including foreign CIDs from forwarded external calls, will be transmitted.
- **Block Foreign CIDs**: This will block any CID that is the result of a forwarded call from off the system. CIDs that are defined for an extension or device will be transmitted.
- **Remove CNAM**: This will remove the CNAM from any CID sent out of this trunk.
- **Force Trunk CID**: This will always use the CID defined for the trunk, except if the trunk is part of an emergency route with an emergency CID defined for the extension or device. Intra-Company routes will always transmit an extension's internal number and name.

- **Maximum Channels**- Controls the maximum number of outbound channels (simultaneous calls) that can be used on this trunk. To count inbound calls against this maximum, use the auto-generated context: from-trunk-[trunkname] as the inbound trunk's context (see extensions_additional.conf). Leave blank to specify no maximum.
- **Disable Trunk**- Check this to disable this trunk in all routes where it is used.
- **Monitor Trunk Failures**- If you have a custom AGI script to call for reporting, logging, emailing or otherwise take some action on trunk failures – you would name it here and check the "enable" box to utilize it.



- **Dialed Number Manipulation Rules**:

  - These rules can manipulate the dialed number before sending it out of this trunk. If no rule applies, the number is not changed. The original dialed number is passed down from the route where some manipulation may have already occurred. This trunk has the option to further manipulate the number. If the number matches the combined values in the prefix plus the match pattern boxes, the rule will be applied and all subsequent rules ignored. Upon a match, the prefix, if defined, will be stripped. Next, the prepend will be inserted in front of the match pattern and the resulting number will be sent to the trunk. All fields are optional.

    Rules:

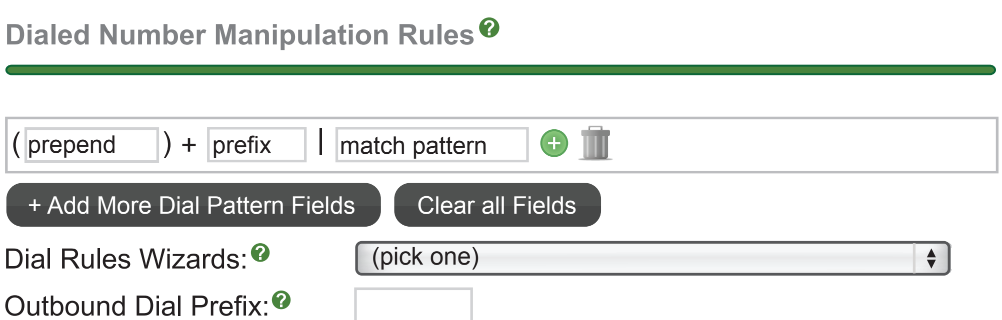
    - **X** matches any digit from 0-9.
    - **Z** matches any digit from 1-9.

- **N** matches any digit from 2-9.
- **[1237-9]** matches any digit in the brackets (example: 1,2,3,7,8,9).
  "." wildcard, matches one or more dialed digits.
- **prepend**: Digits to prepend upon a successful match. If the dialed number matches the patterns in the prefix and match pattern boxes, this will be prepended before sending to the trunk.
- **prefix**: Prefix to remove upon a successful match. If the dialed number matches this, plus the match pattern box, this prefix is removed before adding the optional prepend box and sending the results to the trunk.
- **match pattern**: The dialed number will be compared against the prefix, plus this pattern. Upon a match, this portion of the number will be sent to the trunks after removing the prefix and appending the prepend digits. You can completely replace a number by matching on the prefix only, replacing it with a prepend and leaving the match pattern blank.

◦ **Dial Rules Wizard**- This has the following options.

- **Always dial with prefix**: This is useful for VoIP trunks, where if a number is dialed as "5551234," it can be converted to "16135551234."
- **Remove prefix from local numbers**: This is useful for ZAP and DAHDi trunks, where if a local number is dialed as "6135551234," it can be converted to "555-1234."
- **Setup directory assistance**: This is useful to translate a call to directory assistance.
- **Lookup numbers for local trunk**: This looks up your local number on **www.localcallingguide.com** (NA-only), and sets up so you can dial either 7 or 10 digits (regardless of what your PSTN is) on a local trunk (where you have to dial 1 + the area code for long distance, but only "5551234" (7-digit dialing) or "6135551234" (10-digit dialing) for local calls.
- **Upload from CSV**: Upload patterns from a CSV file, replacing existing entries. If there are no headers, then the file must have 3 columns of patterns in the same order as in the GUI. You can also supply headers: prepend, prefix and match pattern in the first row. If there are less than 3 recognized headers, then the remaining columns will be blank.

◦ **Outbound Dial Prefix**- The outbound dialing prefix is used to prefix a dialing string to all outbound calls placed on this trunk. For example, if this trunk is behind another PBX or is a Centrex line, then you would put "9" here to access an outbound line. Another common use is to prefix calls with "w" on a POTS line that needs time to obtain a dial tone to avoid eating digits. Most users should leave this option blank.



- **Outgoing/Incoming Settings**:
    - **Trunk Name**- Give the trunk a descriptive name such as “mysiptrunk.”
    - **PEER Details**- Here you give the PEER connection parameters supplied by your VoIP provider. You may need to add to the provided default settings and in some cases remove default settings depending on your provider or application.
      WARNING: Order is important, as it will be retained. For example, if you use an “allow/deny” directive, then make sure the “deny” is first (reading top down).
    - **USER Context**- This is most often the account name or number your provider expects.
    - **USER Details**- Here you supply the USER connection parameters supplied by your VoIP provider. You may need to add to the provided default settings and in some cases remove default settings depending on your provider or application.
      WARNING: Order is important, as it will be retained. For example, if you use an “allow/deny” directive, then make sure the “deny” is first (reading top down).

Outgoing Settings

Trunk Name:

PEER Details:

```
host=***provider ip address***
username=***userid***
secret=***password***
type=peer
```

Incoming Settings

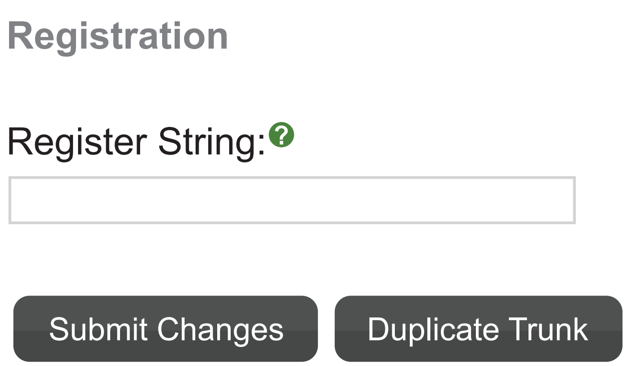
USER Context:

USER Details:

```
secret=***password***
type=user
context=from-trunk
```

- **Register String**- Most VoIP providers require your system to register with theirs. If required, you will need to enter the string the provider specifies, such as **username:password@some.voipprovider.com**. In some cases you may need to provide a DID and the end of the string, such as **username:password@some.voipprovider.com/7045551212**.

- **Submit Changes / Duplicate Trunk**- Press the “Submit Changes” button when finished exiting the trunk and use the “Duplicate Trunk” button to easily create secondary trunks when provided.



Adding a DAHDi Trunk

DAHDi (Digium Asterisk Hardware Device interface) is the driver framework for interfacing with telephony hardware, such as a T1/PRI card. You would refer to the hardware documentation for initial installation. Then turn to the DAHDi Trunk module for integration of the **hardware** into your dial plan (usually in conjunction with the Inbound Route and/or Extensions module. See their respective user guides for more info). There are three core areas displayed for a new DAHDi Trunk: **General Settings**, **Dialed Number Manipulation Rules** and **Outgoing Settings**.

- **General Settings**:

    - **Trunk Name**- Set a descriptive name for the trunk.
    - **Outbound CallerID**- Use this field to specify caller ID for calls placed out of this trunk with the <NXXNXXXXXX> format. You can also use the format: “hidden” <NXXNXXXXXX> to hide the caller ID sent out over digital lines, if supported (E1/T1/J1/BRI/SIP/IAX2).
    - **CID Options**- This setting determines what CIDs will be allowed out of this trunk. Please **NOTE** that Emergency CIDs defined on an extension or device will **ALWAYS** be used if this trunk is part of an emergency route regardless of these settings.

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Intra-Company routes will always transmit an extension's internal number and name.

- **Maximum Channels**- Controls the maximum number of outbound channels (simultaneous calls) that can be used on this trunk. To count inbound calls against this maximum, use the auto-generated context: from-trunk-[trunkname] as the inbound trunk's context (see extensions_additional.conf). Leave blank to specify no maximum.
- **Disable Trunk**- Check this to disable this trunk in all routes where it is used.
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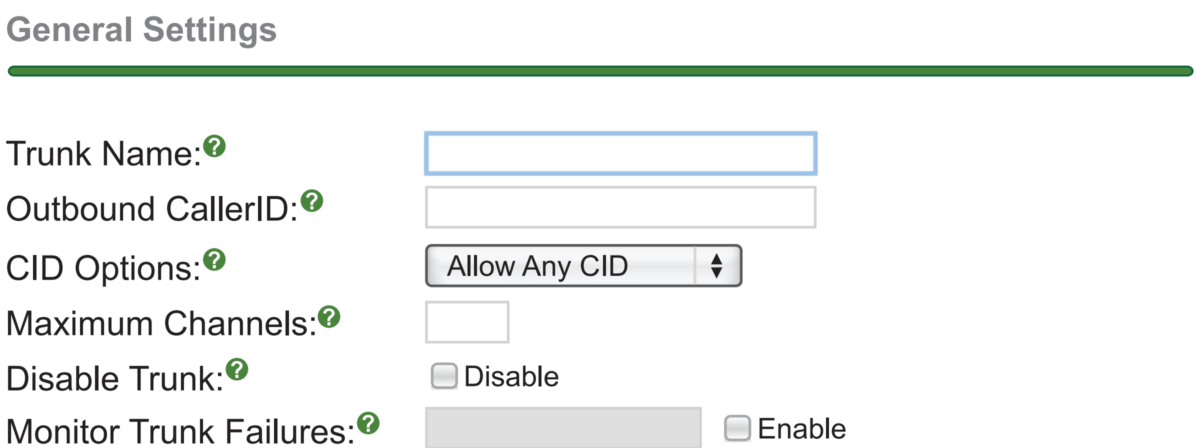

- **Dialed Number Manipulation Rules**:

  - These rules can manipulate the dialed number before sending it out of this trunk. If no rule applies, the number is not changed. The original dialed number is passed down from the route where some manipulation may have already occurred. This trunk has the option to further manipulate the number. If the number matches the combined values in the prefix plus the match pattern boxes, the rule will be applied and all subsequent rules ignored. Upon a match, the prefix, if defined, will be stripped. Next, the prepend will be inserted in front of the match pattern and the resulting number will be sent to the trunk. All fields are optional.

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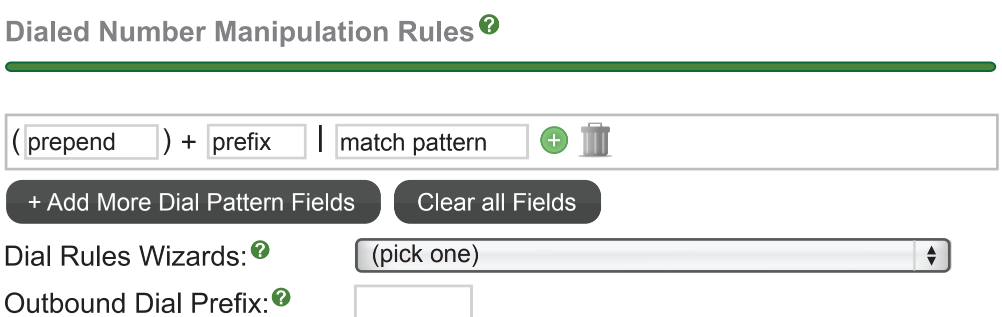


- **Outgoing Settings**:
  - **DAHDi Identifier**- DAHDi channels are referenced either by group or channel number (which is defined in chan_dahdi.conf, often by including dahdi-channels.conf). The default setting is **g0 for group zero**.

Adding an IAX2 Trunk

- IAX2 (Inter-Asterisk eXchange protocol) is a Digium developed VoIP protocol that is extremely useful when you spread your overall dialplan across two or more Asterisk systems.
- Configuration wise, you may refer to the preceding **Adding A SIP Trunk** section, the IAX2 example and **https://wiki.asterisk.org/wiki/display/AST/Home** for general information.

Adding an ENUM Trunk

- ENUM (E.164 Number Mapping) is a method, which uses a special DNS record to translate a telephone number into a URI (Uniform Resource Identifier). URI's in turn can translate into an IP address or a specific extension at a specific IP address or perhaps end up back on the PSTN locally from the receiving system. This facility can be very useful for PSTN unification.
- ENUM lookups are done automatically on the e164.org domain, therefore little configuration is required. Although it is possible to offer outside access to local resources (PSTN) that would be beyond the scope of this guide. See **http://www.e164.org/** for more information.

Adding a DUNDi Trunk

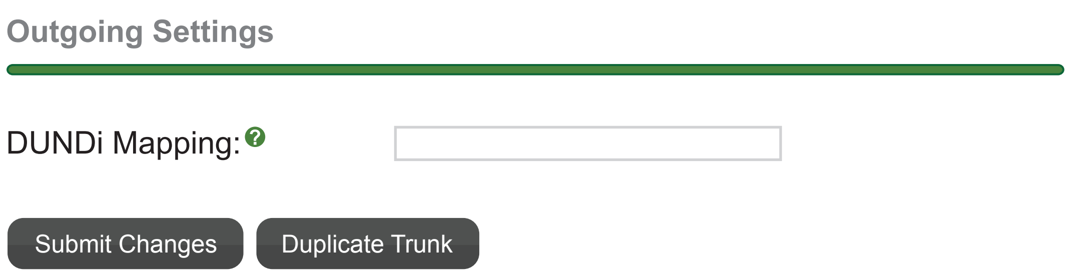
- DUNDi (Distributed Universal Number Discovery) is a trusted, peer-to-peer network that functions somewhat like ENUM, however instead of using centralized service, DUNDi is fully distributed and uses no centralized authority mechanism. See **http://www.dundi.com** for more information and **http://doxygen.asterisk.org/trunk/Config_dun.html** for a dundi.conf example.
- The PBX offers limited support for DUNDi trunks and additional manual configuration is required. The trunk name should correspond to the [mappings] section of the remote dundi.conf systems. For example, you may have a mapping on the remote system and corresponding configurations in dundi.conf locally, that looks as follows:

  [mappings]
  priv => dundi-extens,0,IAX2,priv:${SECRET}@218.23.42.26/${NUMBER},nopartial

  In this example, you would create this trunk and name it priv. You would then create the corresponding IAX2 trunk with proper settings to work with DUNDi. This can be done by making an IAX2 trunk in PBX or by using the iax_custom.conf file.

The dundi-extens context in this example must be created in extensions_custom.conf. This can simply include contexts such as ext-local, ext-intercom-users, ext-paging and so forth to provide access to the corresponding extensions and features provided by these various contexts and generated by PBX.

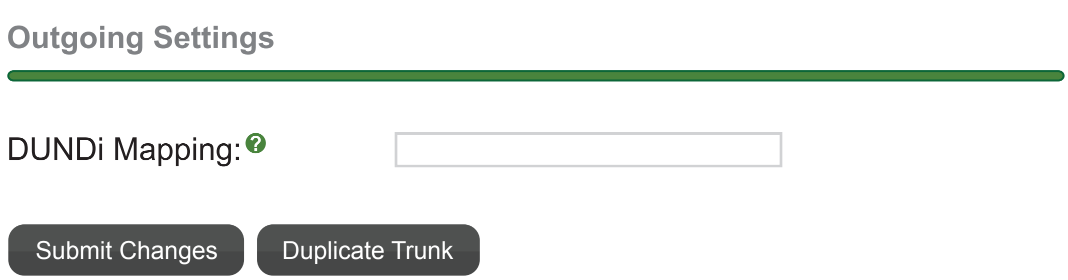
- The **General Settings** and the **Dialed Number Manipulation Rules** are the same as other types of trunks, however the **Outgoing Settings** are unique.
    - **DUNDi Mapping**- This is the name of the DUNDi mappings as defined in the [mappings] section of the remote dundi.conf peers. This corresponds to the "include" section of the peer details in the local dundi.conf file.



Adding a Custom Trunk

- Custom trunks are useful when attepting to integrate certain devices like cellular gateways or perhaps when setting up least cost routing schemas.
- The **General Settings** and the **Dialed Number Manipulation Rules** are the same as other types of trunks, however the **Outgoing Settings** are unique.
    - **Custom Dial String**- Define the custom dial string here. Use the token $OUTNUM$ wherever the number to dial should go. For example:

```
Local/$OUTNUM$@leastcost-custom
SIP/$OUTNUM$@192.168.41.23:5064
IAX2/$OUTNUM$@10.100.1.11
```




```
SIP/$OUTNUM$@XX.XX.XX.XX
```

Recap

- The Trunks module provides as easy to use mechanism for connectivity between your system(s), your VoIP Provider(s) and/or your telephony hardware, therefore the PSTN.

Examples

VoIP services, such as SIPStation™, provides a code, which when entered into the SIPStation™ module, auto-configures not only the trunks, but inbound and outbound routes as well. (See the Inbound and Outbound Routes user guides for more info). All other VoIP providers require manual configuration. Telephony hardware usually entails direct editing of configuration files and possibly other command line level of interaction. Any of these examples may become **obsolete**, so always check the latest from your VoIP or hardware provider. Lastly, these examples do not discuss inbound or outbound routes.

- **SIPStation Trunk**- Note that SIPStation uses redundant trunks, so you would set up an additional trunk named “freepbx2” with a host of “trunk2.phonebooth.net” and “trunk2.phonebooth.net” in the register string. With SIPStation this is automatic.
    - **Trunk Name**: freepbx1
    - **PEER Details**:

```
context=from-pstn
type=peer
insecure=very
qualify=yes
sendrpid=yes
trustrpid=yes
dtmfmode=rfc2833
username=xxxxxxxx
secret=xxxxxx
host=trunk1.phonebooth.net
dissallow=all
allow=ulaw&g729
```

    - **Register String**: username:password@trunk1.phonebooth.net
- **Skype SIP Connect**- This is for inbound calls from the Skype Network.
    - **Trunk Name**: yourSKYPEaccount#
    - **PEER Details**:

```
type=peer
```

```
context=from-trunk
username= yourSKYPEaccount#
secret= yourSKYPEaccounpassword
qualify=yes
insecure=invite
host=sip.skype.com
fromuser= yourSKYPEaccount#
fromdomain=sip.skype.com
disallow=all
allow=ulaw&gsm&alaw
```

- **Register String**: yourSKYPEaccount#:yourSKYPEaccounpassword@sip.skype.com/yourSKYPEaccount#

- **Asterisk Peer to Peer(SIP)**- Note that in this example, the firewall has been setup on both systems with implicit rules allowing this traffic, otherwise the context=from-internal would be a bad move from a security perspective. You would simply replicate this trunk on the remote system, changing only the host= field.

  - **Trunk Name**: FreePBX (or other descriptive name)

  - **PEER Details**:

```
context=from-internal
type=peer
insecure=very
qualify=yes
dtmfmode=rfc2833
canreinvite=no
host=phones.pbxone.net
disallow=all
allow=ulaw&alaw&gsm
```

- **Asterisk Peer to Peer(IAX2)**- The easiest way to illustrate simple IAX2 trunking is with a table.

PBX A w/IP XXX.XXX.XXX.1 Outgoing Settings **Trunk Name:** SeattleOffice **PEER Details:** Host= XXX.XXX.XXX.2 qualify=yes type=peer username=pbxa secret=password	*PBX A w/IP XXX.XXX.XXX.2* Outgoing Settings **Trunk Name:** SeattleOffice **PEER Details:** Host= XXX.XXX.XXX.1 qualify=yes type=peer username=pbxb secret=password
Incoming Settings **USER Context:** pbxb **USER Details:** type=user context=from-internal host= XXX.XXX.XXX.2 secret=password	Incoming Settings **USER Context:** pbxa **USER Details:** type=user context=from-internal host= XXX.XXX.XXX.1 secret=password
Register String: Leave Blank	**Register String:** Leave Blank

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